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Hayward

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(54) **ECHO CANCELLATION WITH DYNAMIC LATENCY ADJUSTMENT**

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This patent is subject to a terminal disclaimer.

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Related U.S. Application Data

(63) Continuation of application No. 09/102,098, filed on Jun. 22, 1998, now Pat. No. 6,563,802.

(51) **Int. Cl.**
H04B 3/20 (2006.01)

(52) **U.S. Cl.** **370/287**; 379/406.01; 455/570

(58) **Field of Classification Search** 370/268, 370/269, 286, 287, 289, 290; 379/406.01, 379/410, 411; 455/570; 375/102
See application file for complete search history.

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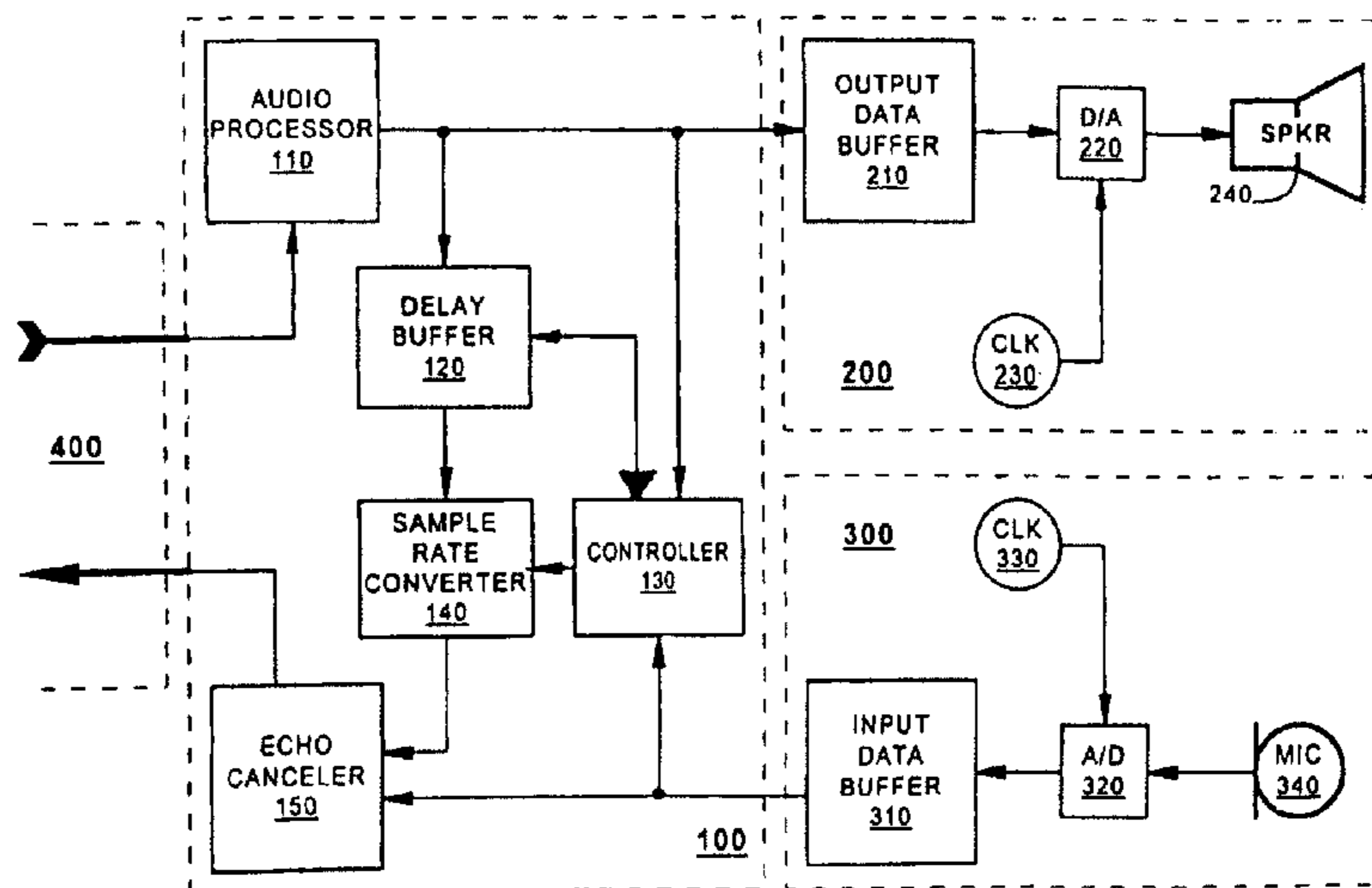
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(57) **ABSTRACT**

An echo cancellation system measures an average delivery rate of a reference data signal and an average capture rate of an input data signal. From the measured data rates, the system converts the reference data signal to a domain of the input data signal. An echo canceler cancels an echo that may be present in the input data signal based upon the converted reference data signal.

14 Claims, 2 Drawing Sheets



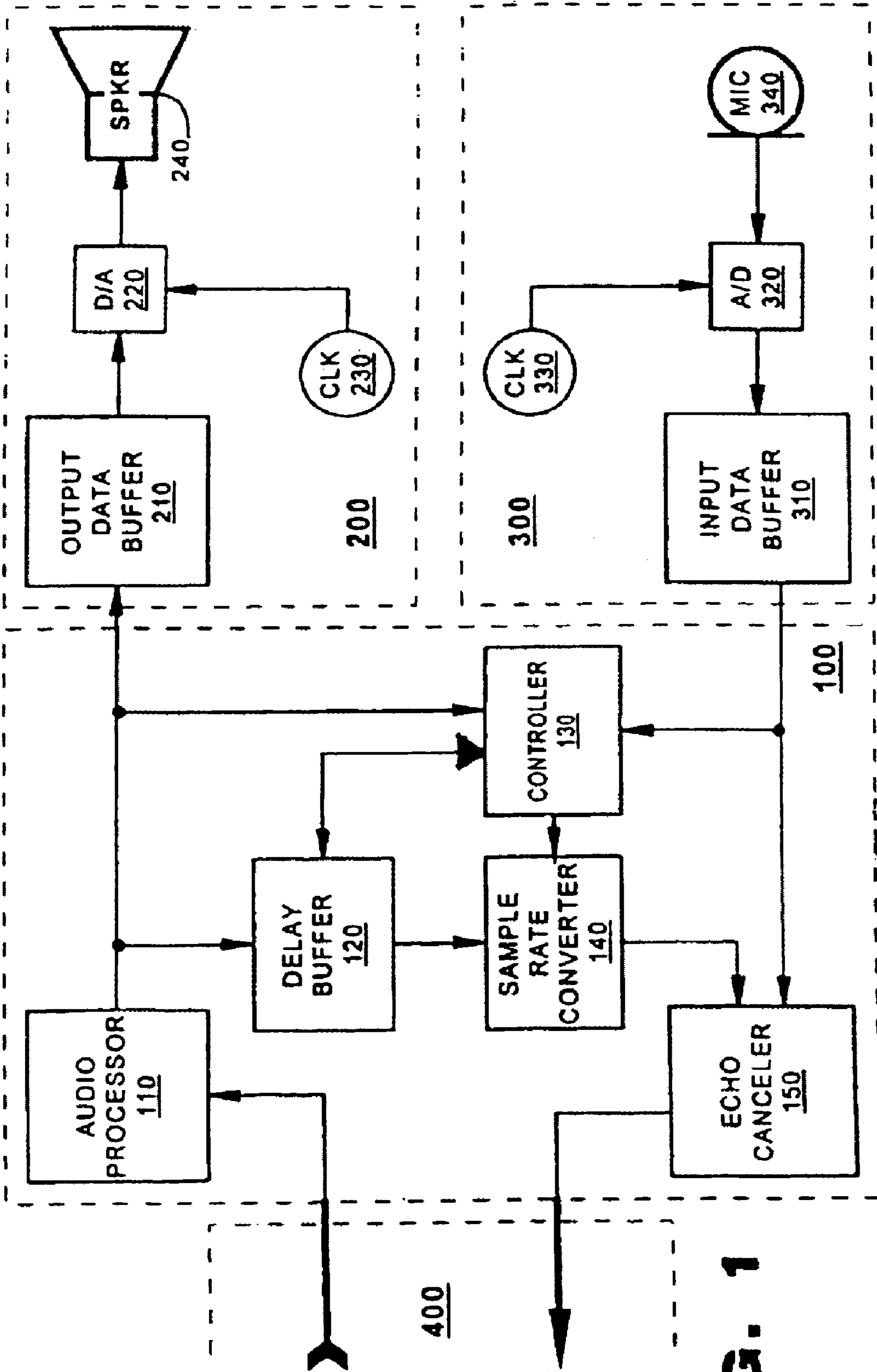


FIG. 1

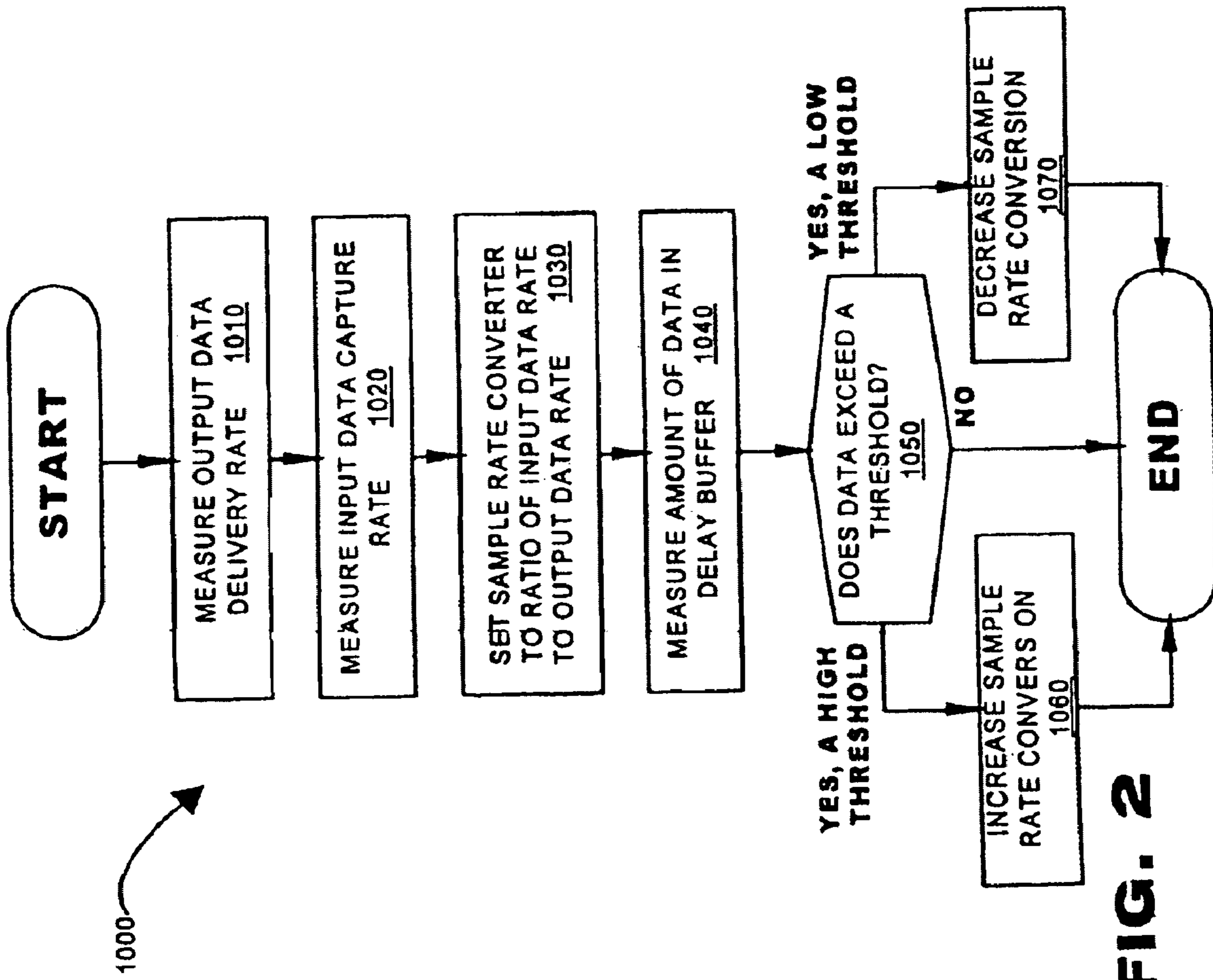


FIG. 2

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ECHO CANCELLATION WITH DYNAMIC
LATENCY ADJUSTMENT

RELATED APPLICATION

This application is a continuation of application titled "Echo Cancellation With Dynamic Latency Adjustment" Ser. No. 09/102,098, filed Jun. 22, 1998 now U.S. Pat. No. 6,563,802.

BACKGROUND

The present invention relates to an improved echo canceler for use with drivers of varying clock rates.

Echo cancellation is known per se. For example, in a speaker phone, echo cancellation prevents sound that is emitted from an omni-directional speaker and captured by a nearby microphone from returning to the signal's source and interfering with communication. Consider an example where a first party speaks to a second party located at the speaker phone. When the first party speaks, the party's voice is broadcast from the speaker phone. Not only is the first party's voice heard by the second party, but the voice also is captured by the speaker phone's microphone. The voice signal reflects off of various surfaces, for example walls, ceilings, furniture and people. The reflected signal is captured by the microphone at some time delayed from the time that the signal was emitted by the speaker. If the reflected signal is not eliminated from the aggregate input signal, the reflected speech signal may be perceived as an annoying echo when delivered to the first party.

Echo cancelers, as the name implies, eliminate the echo generated by the reflected signals. Typically, they do so by buffering a copy of the output audio signal at the speaker phone. The echo canceler monitors the input signal from the microphone and identifies when and how the reflected signal appears in the input signal. When a reflected signal is identified, the processor generates an inverted replica of the reflected signal from the buffered signal and applies it to the input signal. When applied in a correct timing relationship, the replica cancels the reflected signal.

Echo cancelers appear in a variety of applications beyond merely speaker phones. For example, they may be used in video conferencing equipment. In all known echo cancelers, the output speaker equipment and the input microphone equipment are driven by a single clock source. The single clock source permits the correct timing relationship to be maintained between the buffered output signal (the source of the replica) and the captured input signal. Speaker and microphone equipment are not driven by independent clocks because drift among them would prevent the echo canceler from establishing and maintaining the correct timing relationship between the replica and reflected signals.

It is anticipated that computer systems such as personal computers and/or network computers may include hardware that enables telecommunication or video conferencing. However, such computer systems may provide independent speaker and microphone equipment, each with its own clock. Even if the clocks had the same ideal clock rate, echo cancellation heretofore could not be provided for such a system because drift among the two clocks would impair the operation of the echo canceler. Echo cancellation certainly could not be provided for a system where speaker and microphone equipment possessed independent clocks with different clock rates.

Accordingly, there is a need in the art for an echo canceler that is suitable for use with independently clocked input and output devices.

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SUMMARY

The present invention provides improved echo cancellation in which an average delivery rate of a reference data signal and an average capture rate of an input data signal are measured. From the measured data rates, the reference data signal is converted to a domain of the input data signal and input to an echo canceler. The echo canceler cancels an echo that may be present in the input data signal based upon the converted reference data signal.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a block diagram of an echo canceler constructed in accordance with an embodiment of the present invention.

FIG. 2 is a flow diagram of a method of operation of an embodiment of the present invention.

DETAILED DESCRIPTION

Embodiments of the present invention provide an echo canceler that dynamically adapts to clocking differences between input and output drivers. In an embodiment, dynamic latency adjustments are made based upon a first measured rate at which output data is drained from the system and a second measured rate at which input data is captured by the system. By averaging the input and output rates and comparing them, an average skew rate is identified. An adjustable sample rate converter converts a reference signal from a time domain of the output drivers to a time domain of the input driver. The converted reference signal is input to an echo canceler with the captured input signal.

In a second embodiment, the measured rates of data drain and data capture define a target amount of data that should be buffered by the echo canceler. The echo canceler defines high and low thresholds surrounding this target amount. If the actual amount of data that is buffered falls outside of either threshold, the echo canceler adjusts the data rate conversion to compensate.

FIG. 1 illustrates an echo canceling system ("ECS") 100 constructed in accordance with an embodiment of the present invention. The ECS 100 interconnects a speaker system 200 and a microphone system 300 with a main system 400. The main system 400 is the source of data to be output by speaker system 200. The main system 400 also receives data captured by the microphone system 300.

The ECS 100 may be populated by an audio processor 110, a delay buffer 120, a controller 130, a sample rate converter 140 and an echo canceler 150. The audio processor 110 receives a signal, called "the reference signal," from the main system 400. It creates a copy of the reference signal and stores the copy in the delay buffer 120. The audio processor 110 also forwards the reference signal to the speaker system 200 without delay. Optionally, the audio processor 110 may be omitted from the embodiment of FIG. 1. The reference signal will be output by the speaker system 200.

The delay buffer 120 stores the reference signal for later use by the echo canceler 150. A sample rate converter 140 drains the reference signal from the delay buffer 120 at a predetermined rate and converts it from a first time domain, the time domain of the speaker system 200, to a second time domain, the time domain of the microphone system 300. The sample rate converter 140 outputs a microphone-domain representation of the reference signal to the echo canceler 150. The echo canceler 150 also receives a captured input signal from the microphone system 300. Using the microphone-domain reference signal, the echo canceler 150 performs echo cancellation on the captured input signal.

The controller **130** is coupled to an input of the speaker system **200** and an output of the microphone system **300**. The controller **130** observes the transmission of data from the main system **400** to the speaker system **200**. It also monitors transmission of data from the microphone system **300** to the main system **400** (via the echo canceler **150**). Based upon the data rate of output by the speaker system **200** and rate of data capture by the microphone system **300**, the controller **130** determines the differences in clock rates between the speaker system **200** and the microphone system **300**. The controller **130** causes the sample rate converter **140** to implement a conversion rate that reflects the operational differences between the speaker system **200** and the microphone system **300**.

FIG. 2 illustrates a method of operation **1000** of the ECS **100** in accordance with an embodiment of the present invention. The ECS **100** measures an average rate of data delivery to the speaker system **200** (Step **1010**). It also measures an average rate of data capture by the microphone system **300** (Step **1020**). Based upon the rates of data delivery and data capture, the ECS **100** causes the reference signal to be converted to the domain of the input signal (Step **1030**). In a first embodiment, the sample rate converter **140** may be programmed based solely upon the relative input and output data rates.

In a second embodiment, the method **1000** also includes identifying local instability. Based on long-term averages of the output data rates and input data rates, the ECS **100** determines a target amount of data (D) that should be buffered in the delay buffer **120** (Step **1040**). The controller **130** may identify an amount of data actually present in the delay buffer **120** and compare it to the target amount D (Step **1050**). If the true amount of data exceeds the target amount D by more than a predetermined threshold, the ECS **100** may increase the conversion rate applied by the sample rate converter **140** (Step **1060**). If the target amount D exceeds the true amount of data by more than a predetermined threshold, the ECS **100** may decrease the conversion rate applied by the sample rate converter **140** (Step **1070**). And, if the true amount of data is within a predetermined margin established around the target D, no change need be made to the conversion rate.

In an embodiment, the method **1000** may be repeated periodically. For example, the method **1000** may be initiated after observing average rates of data capture and data drain over an initial 2–5 second time period. Based upon the average data rates, the sample rate converter **140** is engaged with an initial rate conversion. By repeating the method **1000** on a periodic basis, say every 30 seconds, the initial rate conversion may be refined.

As is known, clocks exhibit slight fluctuations over long periods of time. Thus a 44.1 KHz clock, may operate at a first, clock rate (say, 44.102 KHz) during a first time interval, than operate at a second, slightly decreased clock rate (say, 44.098 KHz) during a second time interval. Skew between this first clock and a second independent clock, one that exhibits its own fluctuations over time, does not remain constant. Embodiments that periodically repeat the method **1000** automatically account for such fluctuations.

A better understanding of the operation of the present invention may be obtained through an example of the ECS **100** integrated in a specific system. Consider a first example where the ideal clock rates of both the speaker system **200** and the microphone system **300** should be identical, say 44.1 Kilosamples/second (“Ks/s”). However, an acceptable margin for clock error may be $\pm 0.05\%$. Consequently, the two

systems **200, 300** may operate at a relative clock differential of as much as 44 samples/second.

A typical speaker system **200** may include an output data buffer **210**, a digital to analog converter (“D/A”) **220**, a clock generator **230** and a speaker **240**. Data received from the main system **400** is stored in the output data buffer **210**. The D/A **220** drains data from the output data buffer **210** at a rate determined by the clock **230**, converts the data to an analog signal and drives the speaker **240** with it.

A typical microphone system **300** may include an input data buffer **310**, an analog to digital (“A/D”) converter **320**, a clock generator **330** and a microphone **340**. The microphone **340** captures sound and generates an analog signal therefrom. The A/D **320** converts the analog signal to a digital signal at a sampling rate determined by the clock **330**. The digital signal is loaded into the input data buffer **310**. The input data buffer **310** outputs a captured input signal to the ECS **100**.

In the first example above, the speaker system **200** may drain data at 44.122 Ks/s (44.1 Ks/s+0.05%) and the microphone system **300** may capture data at 44.078 Ks/s (44.1 Ks/s–0.05%). If a traditional echo canceler were used, one that includes only a delay buffer **120** and an echo canceler **150**, the clock differential between the two systems **200, 300** would cause reference signal data to be input to the delay buffer **120** at a faster rate than it could be output to the echo canceler **150**. Eventually, the delay buffer **120** would overflow or the echo canceler **150** would lose the necessary timing relationship between the captured input signal and the reference signal stored in the delay buffer **120**. In either case, the echo canceler **150** would cease to function. The echo would remain in the captured signal. By contrast, the conversion rate established by the sample rate converter **140** of the ECS **100** permits echo cancellation to occur despite any operational difference between the clocks **230, 330**.

Consider a second example where the ideal clock rates of the speaker system **200** and the microphone system **300** are different. For example, voice signals in telecommunications applications traditionally are sampled at an 8 Ks/s rate. However, in a given application, audio data may be presented to the speaker system **200** at a second data rate, such as 44.1 Ks/s. A traditional echo canceler could not operate on data signals having markedly different data rates. However, the ECS **100** of the present invention operates successfully on these two data signals. In an embodiment, the sample rate converter **140** may be preprogrammed to convert the reference signal from the ideal 44.1 KHz rate to an ideal 8 KHz rate. Subsequent operation of the method **1000** would refine the conversion rate to actual data rates used by the speaker and microphone systems **200** and **300**.

In practice, particularly where the ECS **100** is used with speaker systems **200** and microphone systems **300** that possess data buffers **210, 310**, exchange of data to and from the main system **400** may occur in high-rate bursts rather than as a continuous stream of data. As is known in computer applications, audio data may be organized into data packets, each packet containing data representing audio over a predetermined period of time (say, 10 ms). Often, a main system **400** multiplexes data exchange with other functions. The packets are stored in the output data buffer **210** and drained from the buffer **210** at a constant rate determined by the clock **230**. Similarly, the microphone system **300** may accumulate captured input data at a steady rate as determined by the clock **330** but may deliver the packets to the main system **400** in high rate bursts.

In a bursty system as described, the clock rates of the speaker system **200** and/or the microphone system **300** are

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not determined from short-term observation of the data delivery rates to or from the main system **400**. Consider an example where packets define audio data for a 10 ms interval. On any given 10 ms interval, the main system **400** may issue a high rate burst of packets that fills the output data buffer **210**. The observed data rate greatly exceeds the rate of clock **230**. However, during subsequent 10 ms intervals, the main system **400** would not issue additional packets to the speaker system **200**. Considered in isolation, no 10 ms interval accurately represents the rate of clock **230**. In a bursty system, the controller **130** considers the average delivery rate of data to the speaker system **200** (and, also an average data rate from the microphone system **300**) to calculate a conversion rate of the sample rate converter **140**. For example, where a data packet defines a 10 ms interval, the traffic controller **130** may identify the average data rates over an interval of 2–5 seconds.

The ECS **100** may be implemented in hardware or software. That is, the audio processor **110**, delay buffer **120**, controller **130**, sample rate converter **140** and echo canceler **150** may be provided as hardware elements in, for example, an integrated circuit such as an application specific integrated circuit. Alternatively, the ECS **100** may be a “software machine,” constituting a general purpose processor or digital signal processor operating according to program instruction. In a software embodiment, FIGS. **1** and **2** illustrate the functionality of the ECS **100**. Thus embodiments of the present invention permit echo cancellation to be performed on signals having different data rates or with input and output drivers that possess independent clocks. It should be appreciated that the ECS **100** of the present invention operates independently of the type of echo canceler **150** that is used. Any of a variety of echo cancelers may be used in the place of echo canceler **150** consistent with the teachings of the present invention.

Several embodiments of the present invention are specifically illustrated and described herein. However, it will be appreciated that modifications and variations of the present invention are covered by the above teachings and within the purview of the appended claims without departing from the spirit and intended scope of the invention.

I claim:

1. A system for echo cancellation, comprising:

a first system input to receive a reference data signal;
a second system input to receive an input data signal;
a controller coupled to the first and second system inputs, wherein the controller to measure a delivery rate of the reference data signal, to measure a capture rate of the input data signal and to dynamically generate a replica of the reference data signal in a time domain of the input data signal based on periodic measurements of the delivery rate and the capture rate; and

an echo canceler to receive the generated replica of the reference data signal and the input data signal.

2. The system of claim **1**, further comprising:

a delay buffer coupled to the first system input.

3. The system of claim **2**, further comprising:

a sample rate converter coupled to the delay buffer and having a control input, wherein the controller having an output coupled to the control input of the sample rate converter.

4. The system of claim **1**, further comprising:

a first clock coupled to the first system input.

5. The system of claim **1**, further comprising:

a second clock coupled to the second system input.

6. A system for echo cancellation, further comprising:

a first system input to receive a reference data signal;

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a second system input to receive an input data signal;

a controller coupled to the first system input and the second system input, wherein the controller to measure an average delivery rate of the reference data signal, to measure an average capture rate of the input data signal, to convert the reference data signal to a time domain of the input data signal based upon the average delivery rate and the average capture rate, to determine a target amount of data that should be buffered based on the average delivery rate and the average capture rate, and when an actual amount of buffered data deviates from the target amount, the controller to adjust a rate of conversion applied when converting the reference data signal; and

an echo canceler to receive the input data signal and the converted reference data signal.

7. The echo cancellation system of claim **6** further comprising:

a delay buffer, coupled to the first system input, to store the reference digital signal.

8. The echo cancellation system of claim **7**, further comprising:

a sample rate converter, coupled to the delay buffer and the controller, to store the converted reference data signal.

9. An echo cancellation system, comprising:

a first system input to receive a reference data signal,

a second system input to receive an input data signal,

a delay buffer coupled to the first system input,

a sample rate converter coupled to the delay buffer and having a control input,

a controller coupled to the first and second system inputs and having an output coupled to the control input of the sample rate converter, wherein the controller to convert the reference data signal to a time domain of the input data signal using a ratio of a measured delivery rate of the reference data signal to a measured capture rate of the input data signal, and

an echo canceler coupled to the second system input and to the sample rate converter.

10. The system of claim **9**, wherein:

the first system input is coupled to an output data device, the second system input is coupled to an input data device, and

the controller adapted to measure rates of data delivery of the reference data signal and the input data signal respectively across each system input.

11. The system of claim **9**, wherein:

the first system input is coupled to an output data device, the second system input is coupled to an input data device, and

the controller to measure average rates of data delivery of the reference data signal and the input data signal respectively across each system input.

12. The system of claim **10**, wherein the output data device and the input data device each include an independent clock.

13. The system of claim **12**, wherein the clocks of the output data device and the input data device have a same ideal clock rate.

14. The system of claim **12**, wherein the input data device and the output data device have different ideal clock rates.

UNITED STATES PATENT AND TRADEMARK OFFICE
CERTIFICATE OF CORRECTION

PATENT NO. : 6,990,084 B2
APPLICATION NO. : 10/288534
DATED : January 24, 2006
INVENTOR(S) : Hayward

Page 1 of 1

It is certified that error appears in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

On the Title page, Item (56), under "U.S. Patent Documents", in column 1, line 1, delete "375/102" and insert -- 375/349 --, therefor.

On the Title page, Item (56), under "U.S. Patent Documents", in column 2, line 20, delete "Banerjca" and insert -- Banerjea --, therefor.

On the Title page, Item (56), under "U.S. Patent Documents", in column 2, line 21, delete "6,268,387" and insert -- 6,266,367 --, therefor.

Signed and Sealed this

Second Day of October, 2007

A handwritten signature in black ink on a light gray dotted background. The signature reads "Jon W. Dudas" in a cursive style.

JON W. DUDAS

Director of the United States Patent and Trademark Office